A NEW SUBBAND ADAPTIVE FILTERING ALGORITHM FOR ANC

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ABSTRACT-The low latency filter system in the active noise control subsystem includes the degradation of input signals and sub-error signals using filter banks and a range of sub-band weights in a full-range noise using a synthetic filter called "stacking". Typically, FIR is a primer for the design of these filtration banks, as well as to minimize the effect of lateral lobe and spectral leakage. The delay in filters is reduced by low pass. The filter design and lateral lobe effect are compensated by the proper staging and accumulation of the sub-weights. Using UDFTM filters, we will increase the number of sub-domains that reduce the workload on the computer but can lead to more interference, thus reducing the performance of the ANC system.
For designing of ANC systems Adaptive Filter place a major role. It reduces the complexity of ANC algorithms, especially when large-scale audio noise and system models have long driving responses. Results showed improved performance and complexity of the new method compared between two sub fields and a block of adaptive filtering algorithms. In this, we proposed a new SAF method based on UDFTM that reduces the side lobe distortion effects and delays provided by the first low-pass typical filter on system performance.

Keywords:**DFT, dolph-linear-phase finite-impulse response, sub band adaptive filtering, SAF, UDFM**

I. INTRODUCTION

SAFtechniques plays a major role in designing of ANCsystems.They decrease the computational complexity of ANC algorithms when the noise is a broad-banded and the system models have a long impulse outputs.ANCis a method of canceling a noise in an acoustic cavity by generates an appropriate anti-noise signal via canceling speakers.

In general, SAF methods provide a good alternative approach to meet ANC requirements, due to their spectral degradation and downstream sampling processes. Because the dynamic spectral domain of the inheritance and the spread of the eigen value of the common signal noise matrix reduces the sub-band performance, ie., the convergence rate, the noise attenuation level, and the stability of the ANC system, improves with SAF techniques. Thus, one expects that increasing the number of sub-domains (or block length) M improves performance.

The delay less SAF scheme in an ANC system involves the decomposition of reference signal(i.e.,input signal) and error signals into sub bands using analysis filter banks, and combining the sub band weights into a full-band noise canceling filter by using synthesis filter bank called "weight stacking". Typically, a linear-phase FIRlow-pass filter (i.e., prototype filter) is designed and modulated for the designing of such filter banks. The filter must be designed in such a way that the side-lobe effect and spectral leakage should minimized. The latter requires a high-order FIR filter, introduce a long delays, which increases with M as the bandwidth shrinks. The side lobe and long delay interference introduced by the prototype filter degrade the performance of SAF algorithms for large M, limiting the computational saving that can be obtained by increasing the number of sub bands.It Improves the system performance and reduce the computational burden by increasing M has inspired the work presented herein.

II. LITERATURE REVIEW

2.1 Low-Resource Delay less Sub band Adaptive Filter Using Weighted Overlap-add

A delay less structure for low-resource implementation is proposed to eliminate filter bank processing delays in SAFs. Rather than using direct IFFT or poly phase filter banks to transform the SAFs back into time-domain, the proposed method utilizes a "weighted overlap-add" (WOLA) synthesis. Low-resource real-time implementations are targeted and do not involveas long as the echo plant FFT or IFFT operations. Also, the proposed method facilitates time distribution of the adaptive filter reconstruction calculations crucial for efficient real-time and hardware implementation. This method is implemented on an over-sampled WOLA filter bank employed as part of echo cancellation application. Evaluated results shows that the new method performs conventional SAF systems since the signals used in actual adaptive filtering are not distorted or deformity by filter bank aliasing. This method is a good match for partial update adaptive algorithms since the segments oftime-domain adaptive filter are sequentially reconstructed and updated.

2.2 Adaptive Feedback Active Noise Control Basics

A delay less method is proposed by SAF for adaptive filtering technique. This method is mainly based on WOLA synthesis of the SAFs,and is very desired and is well mapped to a less-resource hardware implementation. The performance of an open-loop version of system was compared against a conventional SAF system employing the same WOLA analysisfilter banks, with the proposed delay less system offering superior performance but at high computational cost. The performance is similar to the DFT-FIR delay less SAF system that employs straightforward poly phase filter banks.

However, the WOLA-based SAF synthesis offers a superior mapping to low-resource hardware with limiting the precision arithmetic. Also the WOLA adaptive filter reconstruction may be easily spread out in time simplifying the necessary hardware. This time-spreading may easily combined with the partial update adaptive algorithms for reducing the computational cost for low-resource real time platforms.It will discuss about various method of noise reduction for wireless communication.In any communication system Noise is an undesiredsignal and inevitable interference. It is non-informative and plays the role of destroying the intelligence of the reference signal. A signal traveling through the channel also gets lots of noise**.** It degrades the original signal quality**.** The effect of noise could be reduced only by using proper bandwidth of the channel, which is again undesired, as bandwidth is a precious resource. Hence to regenerate original signal, it reduces the efficiency of noise signal, or in the other way, raise the original signal efficiency level, at the receiver and also improves the SNR.

III. DELAYLEESS SUBBAND ADAPTIVE FILTER ALGORITHM METHODS

Less delay includes sub-band Adaptive Adaptive Technology 1) Full-range filter filters the bookmark. 2) Analyze input and error signals in subdomains. 3) Loss (decrease) in the subdomains. 4) Adaptive filters are used in subdomains. 5) The weighting method is used to integrate all weights of subdomains into the full range filter. The signal is analyzed by synthesis synthesis banks. Adaptive filters are used in subdomains and all sub-band weights are aggregated together to make the noise canceling filter fully. The sub-scale weights are stacked by synthetic filters. The analysis and synthesis filter banks must be designed in such a way that they make a perfect reconstruction pair [1]. Fig. 1 shows the delayless subband adaptive filtering used in the ANC FxNLMS algorithm [2]. In this method,x(n) is filtered by ŝ(z), producex`(n). x`(n) and e(n) are then filtered into M subbands named x`k(n) and ek(n) using the analysis filter bank h(z) with a decimation(decreasing) factor D. Filter h(z) is given by

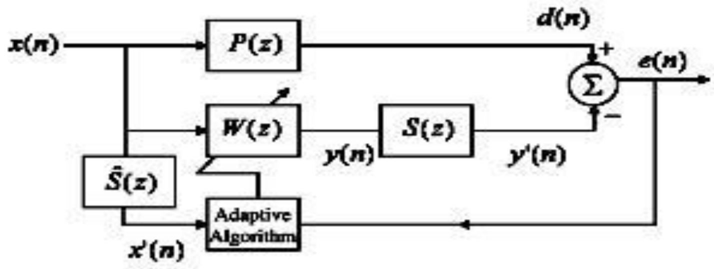


Fig.1

$$h(z) = [H_0(z), H_1(z)........ H_{M-1}(z)]^T$$

All w_K^{SAF} are stacked together to construct w by employing weight stacking methods. The architecture is delayless since y(n) is not explicitly formed as a combination of subband adaptive filters as outputs. The analysis filter were developed is mainly based on uniform DFT modulated (UDFTM) filter banks [2] and tree structured filter banks like Hadamard transform [3]. Several methods such as fast Fourier transform (FFT)-1 [2], FFT-2 [4], DFT-FIR [4], and linear weight transform [5], have been suggested for "weight stackingUDFTM filter are composed a set of bandpass filters created by modulating a linear phase lowpass filter. The central frequencies of the bandpass filters are at $w_K = 2\Pi k/M$, for 0<k<M-1 and the low pass filter has a bandwidth of Π/M . To exploit the computational advantage of the FFT algorithm, usually M and the length of the low-pass filter are chosen in the powers of 2. The spectral leakage in the analysis filter banks is minimized when the prototype low-pass filter has a very small stop-band energy [1] and a linear phase filter with an even-symmetric impulse response [6]. To decrease the stop-band energy for a bandwidth of Π/M, the filter order should increase. The frequency responses of such prototype LPF designed for lengths of M,2Mand 4M for M=16 by using the Remez algorithm. ALPF of length 4M is needed [2]. Since the prototype LPF should have linear phase, its inherent delay increases with M, changes the executionof the SAF algorithm [7], [8]. The effect of delay is more when the length of w is increased for primary and secondary paths with long impulse responses. Since long w requires a lower step size, the adaptive system becomes more sensitive to noise caused by spectral leakage and delay [8]. The LPF designing is an optimization problem that jointly decreases the delay and stop-band energy. These two phenomenon are inversely related, i.e., increasing the length of the LPFdecreases the leakage of spectrum and increases the delay and vice versa. The methods used to design such LPF are based on quadrature optimization [6], min-max optimization [9], [10], least square[11], and homomorphic filtering [12], [13].

The following notation is used in the project.

NAL	Noise attenuation level
n	Time(sample) index
M	Number of subbands
k	Subband index,0 $\leq$ k $\leq$ M-1
D	Decimation factor used in analysis
Lp	Length of prototype low-pass filter
w	weight vector(coefficients) of the FIR noise canceling filter W(z)
N	Length of **w**
$\mathbf{w}_k^{SAF}$	weight vector for the kth subband filter
L_{SAF}	Length of each subband adaptive filter

W(z)	z-transform of **w**
$\mathbf{w}_k^{SAF}(z)$	z-transform of $\mathbf{w}_k^{SAF}$
$((.))_D$	Modulo-D operation.

Bold upper case and bold lower case letters denote vectors and matrices respectively.

A new SAF algorithm based on UDFTM filter banks aregiven. The preferred low-pass filter in the UDFTM filter banks is of length M and delay $(M-1)/2$ and hence, introduces side lobe and less delay attenuation than other cases, as shown. As will be shown, the side-lobe effect is reduced by over-sampled in subbands and proper weight stacking of the subbands. Unlike existing BAF and SAF algorithms, the proposed SAF algorithm improves the working of the system and reduces the calculation complexity M as increases.

3.1 ALGORITHMS

There are different types of adaptive filtering algorithms:

1.The least squares algorithm (LMS)

2. Normalize the lower average square algorithm (NLMS)

3. Variable size step LMS (VSLMS) algorithm

4.Variable step size algorithm for LMS (VSNLMS) normalization

5. Recursive least squares algorithm (RLS).

3.2 COMPARISON OF ADAPTIVE FILTERING ALGORITHMS

Algorithm: LMS Algorithm

Average attenuation: -18.2 dB

Multiplication operations: 2N+1

Comments: It is veryeasy to implement and is stable when the step size parameter is selected appropriately. This requires knowledge of the input signal prior which is not feasible for the echo cancellation system.

Algorithm: NLMS Algorithm

Average attenuation: -27.9dB

Multiplication operations: 3N+1

Comments: Easyto implement and computationally efficient. Shows very good attenuation and changing step size allows stable performance with non-stationary signals. This was the best choice for real time implementation.

Algorithm: VSSLMS Algorithm

Average attenuation: -9.8 dB

Multiplication operations: 4N+1

Comments: This algorithm provides very poor performance, possibly due to non-stationary nature of speech signals. Only half the attenuation of the standard LMS algorithm takes place. This is not considered for real time implementation.

Algorithm: VSSNLMS Algorithm

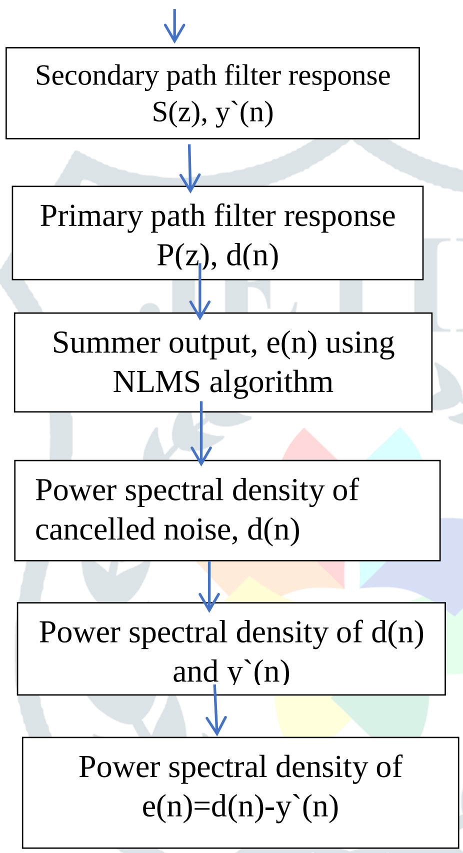
Average attenuation: -9.9 dB

Multiplication operations: 5N+1

Comments: If there is increase in multiplications it gives negligible improvement in performance over VSSLMS algorithm.

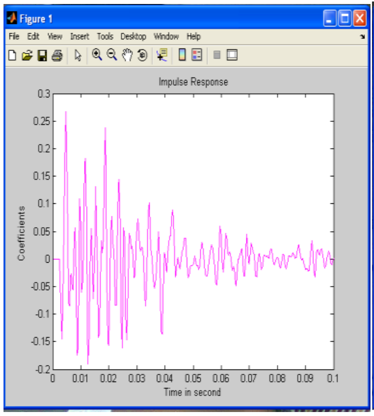
3.3 DATA FLOW DIAGRAM

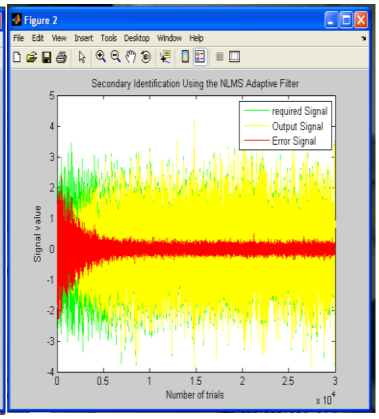
NOISE



IV. RESULTS AND DISCUSSION

The input file 'file1.wav' is read by the command waveread and impulse response of secondary path is plotted as shown in fig 4.1. This means the output of secondary path y`(n) initially. Then estimating the secondary propagation path S^(z) and identifying this path using NLMS adaptive filter. This is shown in fig 4.2 and also shows the required signal d(n),output signal y`(n) and error signal e(n).If the number of iterations are increased, the error signal e(n) is reduced. Here d(n) and y`(n) are having same signal value from $0.5*10^4$ to $3*10^4$.

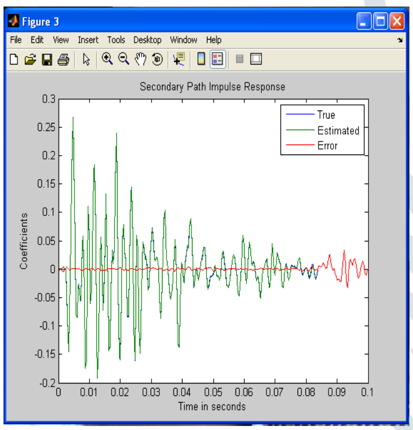




*4.1)Secondary path filter response*4.2)*Secondary path identification Using NLMS*

Fig4.3shows accuracy of the estimated secondary propagation path S^(z).Also the summer output e(n) in time-domain. Here the d(n) and y`(n) follows the same path and error signal e(n) is zero.

Fig 4.4 shows the primary propagation path filter response P(z).The output of P(z) is d(n),this is the actual noise to be canceled by generating anti-noise through the S^(z).Then the noise in the system is to be canceled.



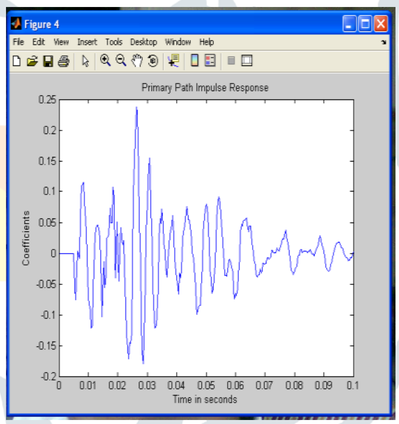
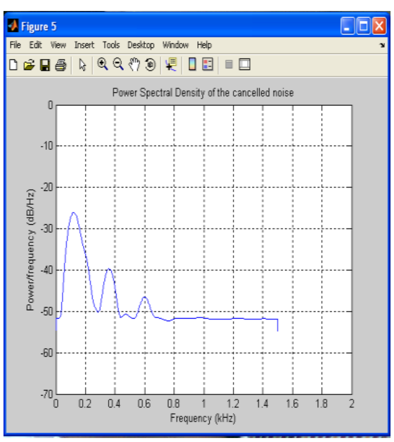


Fig 4.3 Accuracy of Secondary path *Fig 4.4 Primary path filter response*



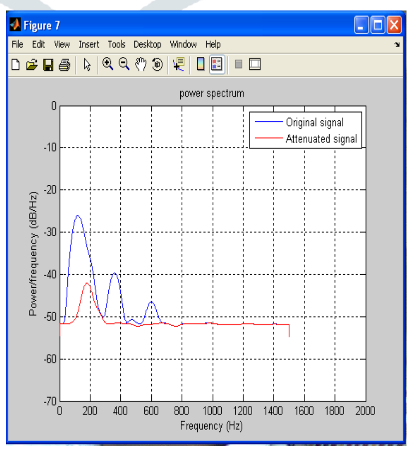


Fig 4.5 Power spectral density of canceled Noise *Fig 4.6 Residual error signal spectrum*

fig 4.5 shows the power spectral density of the canceled noise d(n). Power spectral density means the distribution of d(n) over frequency-axis. The voice frequency range is considered as from (0.3-3.5) KHz.

Fig 4.6 shows residual error signal spectrum of e(n) or power spectral density of d(n) and y`(n).From (0-1300)Hz, there is small difference between d(n) and y`(n).After that both are equal.

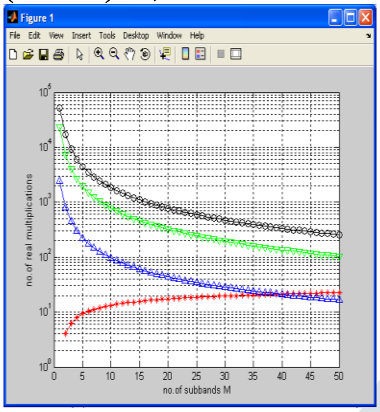


Fig 4.7: Comparison of computational

complexity per input sample

versus number of subbands (M)

Fig 4.7 shows the power spectral density of d(n)-y`(n).At 200Hz, power/frequency is -50 db/Hz (0.0031) approximately.

V. CONCLUSION

Acoustic paths are those encountered in ANC application usually have long impulse responses, which require larger adaptive filters for noise cancellation. Subband adaptive filters working with a large number of subbands have been a very good solution to this problem. The focus of this project was to design a high-performance SAF algorithm. The performance limiting factors of existing SAF structures were found to be due to the inherent side lobes and delays of the prototype low pass filter in the analysis filter banks. Hence, the analysis filter banks were modified to reduce the delay. A new weight stacking transform was designed to alleviate(decrease) the interference introduced by the side-lobes. The modifications resulted in a new subband method that improves the performance and reduces the computational complexity for a larger number of subbands.

The new technique works very well with a larger number of subbands, improves the system performance and attaining lesser complexity, whereas the MT method fails to converge and the performance of the DFT-MDF method deteriorates(decays).

ACKNOWLEDGEMENT

We are grateful for the financial support for this work provided by the JYOTHISHMATHI INSTITUTE OF TECHNOLOGY AND SCIENCE and all faculty members of the ECE department to carry out this work successfully.

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